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10/597,215	07/17/2006	Yutaka Banba	40872	7852

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EXAMINER	
KAZEMINEZHAD, FARZAD	

ART UNIT	PAPER NUMBER
2626	

NOTIFICATION DATE	DELIVERY MODE
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Notice of the Office communication was sent electronically on above-indicated "Notification Date" to the following e-mail address(es):

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Office Action Summary	Application No.	Applicant(s)	
	10/597,215	BANBA, YUTAKA	
	Examiner	Art Unit	
	FARZAD KAZEMINEZHAD	2626	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE _3_ MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☐ Responsive to communication(s) filed on _____.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) _1-14_ is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) _1-14_ is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☒ The specification is objected to by the Examiner.

10)☒ The drawing(s) filed on _____ is/are: a)☒ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☒ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☒ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☒ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _11/2/2006, 7/17/2006_.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application
6) ☐ Other: _____.

DETAILED ACTION

Priority

1. Acknowledgment is made of applicant's claim for foreign priority under 35 U.S.C. 119(a)-(d).

Information Disclosure Statement

2. The information disclosure statement (IDS) submitted on 7/17/2006 mailing date. The submission is in compliance with the provisions of 37 CFR 1.97. Accordingly, the information disclosure statement is being considered by the examiner.

Specification

3. The disclosure is objected to because of the following informalities: The title is not sufficiently descriptive of the claimed material.

Appropriate correction is required.

Claim Rejections - 35 USC § 103

4. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

Claims 1-14 are rejected under 35 U.S.C. 103(a) as being unpatentable over Imai et al. (US Patent 5,864,800) in view of Yin (US Patent 6,104,996), and further in view of Wu et al. (US 2002/0116199).

Regarding claim 1, Imai et al. does teach an audio signal encoding method, comprising: a producing step of dividing an audio signal into a plurality of sub-band signals (Abstract, Col. 1 lines 15-18 teach dividing a frequency spectrum (e.g. a band) into plural sub-bands; and Col. 1 lines10-11 disclose audio signals as what the invention is directed to), sampling said sub-band signals at respective down-sampling rates depending on the number of said divided sub-band signals, and producing down-sampled sub-band signals (Col. 10 lines 54-57 referring to Fig. 10 teach an example of a band splitting of an incoming signal into two sub-bands (S_A and S_B+S_C respectively) where according to Col. 11 lines 12-17 the down sampling circuit module 39 in Fig. 10 converts the incoming sampling frequency of 88.2 KHz into S_A which has one-half (i.e. one over the number of sub-bands (implying the down sampling frequency depends on the number of the sub-bands)) the frequency of the incoming signal; Col. 11 lines 4-10 also teach this down sampling is accompanied by a bit reduction in the sub-bands by the Bit Reduction module unit 12);

Imai et al. does not teach an encoding step of quantization of said down-sampled sub-band signals, said encoding step being of calculating a linear predictive coefficient from a previously decoded signal on the basis of a backward adaptive prediction method.

Yin does teach an encoding step of quantization of said down-sampled sub-band signals (Col. 7 lines 50-54 teach the Dynamic Bit Allocator unit 700 in Fig. 1 to quantize sub-bands generated from a band division of an incident audio signal by the method steps outlined in the flow chart of Fig. 3 step 308),

said encoding step being of calculating a linear predictive coefficient from a previously decoded signal on the basis of a backward adaptive prediction method (Col. 4 line 67 and Col. 5 lines 1-3 teach the sub-band signals (104) are sent into backward linear predictor modules 400 in Figs. 1 and 2 to undergo back ward linear predictor process. Col. 5 lines 49-55 including Eq. 1 teach linear predictor corrector (LPC) coefficients used to represent the predictor signal in Eq. 1 where the coefficients are time dependent and thereby adaptive; Backward LPC coefficients depend on reproduced decoded signals of past sub-frames of a given frame; i.e. only audible differences are encoded) .

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made that utilizing the Dynamic Bit Allocator (unit 700 Fig. 1) and quantizer (unit 600 in Fig. 1) and its associated methods in Fig. 3 step 308 of Yin downstream of the down sampling unit 39 in Fig. 10 of Imai et al. would enable the latter to quantize its sub-bands and perform backward linear prediction on the incoming signal and thereby compress an audio signal and decrease data redundancy for easier transmission.

Imai et al. in view of Yin do not teach producing vector indexes from down-sampled sub-band signals by performing vector quantization on the basis of an analysis-by synthesis.

Wu et al. does teach utilizing vector quantization as a method of data compression and decompression (¶ 0005 lines 1-6; ¶ 0011 lines 1-4). ¶ 0012 lines 1-6 teach the compression and decompression is made possible by a novel boundary analysis and synthesis framework (i.e. the quantization causing the compression and decompression is based on an analysis by synthesis method) to minimize the data block quantization induced discontinuity. ¶ 0106 does teach the quantization function (unit 108 Fig. 2) to be responsible for carrying out the adaptive sparse vector quantization.

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made that utilizing the quantization module (unit 108) in Fig. 2 of Wu et al. into the systems of Imai et al. in view of Yin by inserting it downstream of the down sampling unit 39 in Fig. 10 of Imai et al. would enable the latter to vector quantize the down sampled sub-band signal based on an analysis by synthesis method and therefore achieve far better compression attributed to the vector quantization as opposed to scalar quantization of Yin and also lead to a smooth signal without substantial discontinuities at the sub-band boundaries.

Regarding claim 2, Imai et al. does teach the audio signal encoding method as set forth in claim 1; Imai et al. does not teach in his method said encoding step is of producing an excitation vector by using the addition of at least two vector code books.

Wu et al. does teach encoding step is of producing an excitation vector by using the addition of at least two vector code books (¶ 0005 lines 6-9 teach each vector quantization typically searches a codebook which can match the input vector most closely or each vector quantization is associated with a codebook; ¶ 0014 lines 5-13 teach in one embodiment in order to achieve zero-latency, a first quantization (utilizing one codebook) is followed by a second quantization (utilizing another codebook) which is intended to reduce the errors due to the first quantization; finally using the first and second quantization indices (i.e. attributed to two vector codebooks) the final encoded indices are formed as a bit-stream).

For obviousness analysis please see claim 1.

Regarding claim 3, Imai et al. does not teach the audio signal encoding method as set forth in claim 1, in which said encoding step is of calculating, as a difference signal, the difference between a predictive excitation gain and a real excitation gain and performing adaptive scalar quantization of said difference signal.

Yin in Col. 9 lines 15-25 referring to Eq. 11 teaches computing prediction gain (G_l) using the actual sample and the predicted sample data which itself depends on backward adaptive linear prediction coefficients as shown in Eqs. 1-3 in Col. 5 lines 50-65. Col. 6 lines 32-35 teach the linear prediction method and therefore the prediction

samples utilized to be of the type “Code Excited Linear Prediction” type and therefore the prediction gain (G_l) to correspond to predictive excitation gain. In Eq. 3 also it computes the difference between the prediction gain, G_l (predictive excitation gain) and G_(previous) which is itself the total gain from previous block (or previous frame) which corresponds to the real excitation gain. Furthermore the prediction sample used in Eq. 11 according to Eqs. 1-3 (Col. 5 lines 55-65) depends on the quantized signal which is of scalar type because it encodes data points individually and therefore Eq. 13 describes the difference between code excited prediction gain and real excitation gain utilizing backward linear prediction methods with scalar quantization.

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made that utilizing the Dynamic Bit Allocator (unit 700 Fig. 1) and quantizer (unit 600 in Fig. 1) and its associated methods in Figs. 3-5 of Yin downstream of the down sampling unit 39 in Fig. 10 of Imai et al. would enable the latter to quantize its sub-bands and perform backward linear prediction and thereby compute the difference in the prediction and previous excitation gains and thereby only transmit the bits corresponding to the excitation gains and not the entire signals from one frame to the next.

Regarding claim 4, the first limitation (excluding the preamble treated below) is identical to claim 1 and is therefore rejected under similar rationale.

Imai et al. does teach an audio signal decoding method of decoding an audio signal encoded on the basis of an audio signal encoding method and a synthesizing

step of interpolating said reproduced sub-band signals at respective up-sampling rates, and reproducing said audio signal from said interpolated sub-band signals, said decoding step being of calculating a linear predictive coefficient from a previously decoded signal on the basis of said backward adaptive prediction method (This is just the obvious inverse of the coding method, because for every encoding module (e.g analysis filter bank and down-sampling modules) there exists corresponding decoding modules such as the band synthesis unit 52 according to Col. 15 lines 28-38 referring to Fig. 18 and in another embodiment Col. 15 lines 44-57 referring to Fig. 19 teach up-sampling unit 56 to raise the sub-band frequency prior to synthesis);

Imai et al. does not teach said audio signal decoding method comprises a decoding step of reproducing said down-sampled sub-band signals from said vector indexes by performing the inverse vector quantization of said vector indexes.

Wu et al. does teach an inverse vector quantization module in which quantization indices are converted to signal coefficients (¶ 0146 lines 1-3 referring to module 204 in Fig. 3).

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made that utilizing the inverse quantization module 204 in Fig. 3 of Wu et al. upstream of (before) the Up-Sampling module of Wu et al. (unit 56 in Fig. 19) will enable the latter to de-quantize the incoming vector quantized bit streams associated with the sub-bands prior to their up-streaming and synthesis into audio signals and thereby complete the cycle of smoothing initiated with the original

quantization and result in smooth signals without substantial discontinuities at the sub-band boundaries.

Regarding claim 5, Imai et al. does teach audio signal decoding method as set forth in claim 4, in which said decoding step is of receiving said encoded signal is on the basis of said audio signal encoding method (Fig. 1B teaches a schematic diagram of the decoder from a signal which was encoded utilizing its associated (i.e. based on the encoding of the) encoder in Fig. 1A);

Imai et al. does not teach his methods in which said encoding step is of producing an excitation vector by using the addition of at least two vector code books, and said decoding step is of producing an excitation vector by using the addition of at least two vectors equivalent to said vector indexes.

Wu et al. does teach his methods in which said encoding step is of producing an excitation vector by using the addition of at least two vector code books, and said decoding step is of producing an excitation vector by using the addition of at least two vectors equivalent to said vector indexes (¶ 0005 lines 6-9 teach each vector quantization typically searches a codebook which can match the input vector most closely or each vector quantization is associated with a codebook; ¶ 0014 lines 5-13 teach in one embodiment in order to achieve zero-latency, a first quantization (utilizing one codebook) is followed by a second quantization (utilizing another codebook) which is intended to reduce the errors due to the first quantization; finally using the first and second quantization indices (i.e. attributed to two vector codebooks) the final encoded

vector indices are formed as a bit-stream; ¶'s 0136 and 0137 referring to the audio decoding and bit-stream decoding teach that any bit-stream previously generated by the encoder (this will include bit-stream indices attributed to adding two indices corresponding to two vector code books) does undergo bit-stream decoding by the module 200 in Fig. 2 producing the corresponding signal (excitation vector)).

It would have therefore been obvious to one with ordinary skill in the art at the time the invention was made that utilizing the bit stream decoding unit 200 and the inverse quantization module 204 in Fig. 3 of Wu et al. upstream of (before) the Up-Sampling module of Wu et al. (unit 56 in Fig. 19) will enable the latter to de-quantize the incoming vector quantized bit streams attributed to the two quantization procedure above resulting in decoded signals with lower error rates.

Regarding claim 6, Imai et al. does teach the audio signal decoding method as set forth in claim 4, in which said decoding step is of receiving said vector indexes encoded on the basis of said audio signal encoding method (Fig. 1B teaches a schematic diagram of the decoder in which the bit stream (comprising vector indices) from a signal which was encoded utilizing its associated (i.e. based on the encoding of the) encoder in Fig. 1A is being received for decoding process)

Imai et al. does not teach his methods in which said encoding step is of calculating, as a difference signal, the gain difference between a predictive excitation gain and a real excitation gain

and said decoding step is of calculating, as an excitation gain, the addition between said predictive excitation gain and said gain difference obtained from said quantized difference signal on the basis of said backward adaptive prediction method;

Yin does teach his methods in which said encoding step is of calculating, as a difference signal, the gain difference between a predictive excitation gain and a real excitation gain, and performing the adaptive scalar quantization of said difference signal (please see claim 3 treatment above)

and said decoding step is of calculating, as an excitation gain, the addition between said predictive excitation gain and said gain difference obtained from said quantized difference signal on the basis of said backward adaptive prediction method (this comprises the obvious decoding of the previous encoding step; Yin Col. 11 lines 60-64 referring to Fig. 7 teach one decoding module per every encoding module of Fig. 2 and Col. 12 lines 2-4 teach same algorithms are used for the decoding as for the audio encoding in backward predictor including prediction gain calculations and contributions);

For obviousness analysis please see claim 3.

Regarding claim 7, Imai et al. does teach in a transmitter comprising an encoding unit for encoding an audio signal on the basis of an audio signal encoding method which comprises a producing step of dividing said audio signal into a plurality of sub-band signals, sampling said sub-band signals at respective down-sampling rates depending on the number of said divided sub-band signals, and producing said sub-band signals

sampled at said down-sampling rates, and an encoding step of producing vector indexes from said down-sampled sub-band signals by performing the vector quantization of said down-sampled sub-band signals on the basis of an analysis- by-synthesis method, said encoding step being of calculating a linear predictive coefficient from a previously decoded signal on the basis of a backward adaptive prediction method (this limitation is identical to claim 1 when used in a transmitter and is therefore rejected under similar rationale; Imai et al. Col. 1 lines 9-11 specifically disclose the methods of the invention to be directed for transmitting digital signals and the Abstract teaches the system and its respective methods for the various embodiments such as the transmission system in Fig. 22),

said transmitter is adapted to transmit said audio signal encoded by said encoding unit (Imai et al. Fig. 1A shows generation of bit streams ready for transmission and Fig. 22 depicts a transmission system transmitting encoded signals),

Wherein said encoding unit includes an audio signal dividing filter bank (Imai et al. Col. 10 lines 40-45 referring to Fig. 8 teach an incoming signal being divided by an analysis filter bank module 36)

for dividing said audio signal into a plurality of sub-band signals, sampling said sub-band signals at respective down-sampling rates depending on the number of said divided sub-band signals, and producing said sub-band signals sampled at said down-sampling rates, and an encoder for producing vector indexes from said down-sampled sub-band signals by performing the vector quantization of said down-sampled sub-band signals on the basis of an analysis-by- synthesis method, said encoder being adapted to

calculate a linear predictive coefficient from a previously decoded signal on the basis of a backward adaptive prediction method (this limitation is identical to claim 1 and is therefore rejected under similar rationale).

Regarding claims 8 and 9, depending on claim 7, the claim limitations correspond to the method limitations of the method claims 2 and 3 respectively and are therefore rejected under similar rationale.

Regarding claim 10, the first limitation corresponds to the systems corresponding to the methods of the method limitations of claim 4 implemented on a receiver and is therefore rejected under similar rationale. The second limitation corresponds to the second limitation of claim 4 when a synthesis filter bank is utilized. Note that Imai et al. Col. 14 lines 48-53 referring to Fig. 16 teaches receiver capability in the module unit 47 and in Fig. 22 it teaches domains near a transmitter station in which different types of sub-bands may be received. Furthermore Imai et al. Col. 15 lines 32-36 referring to Fig. 18 specifically discloses a synthesis filter bank (unit 56) utilized to restore sub-bands to their original band width.

Regarding claim 11, it teaches decoding of a signal after teaching its encoding by the method of the claim 2. The encoding part is identical to method claim 2 and is therefore rejected under similar rationale. Imai et al. does teach a decoding procedure for every encoding procedure taught, such as Fig. 1B taking in encoded bit streams

produced in Fig. 1A for decoding and Fig. 16 utilizing a synthesis filter bank (unit 56) for adding sub-bands produced by an analysis filter bank (unit 36 in Fig. 8) in its corresponding encoding taught in Fig. 8.

Regarding claim 12, depending on claim 7, it corresponds to the systems corresponding to the method limitations of the method claims 3 and 6 on a receiver and is therefore rejected under similar rationale. Note that Imai et al. Col. 14 lines 48-53 referring to Fig. 16 teaches receiver capability in the module unit 47 and in Fig. 22 it teaches domains near a transmitter station in which different types of sub-bands may be received.

Regarding claim 13, its limitations are identical to the system (transmitter) limitations of claim 7 and is therefore rejected under similar rationale. Note that the transmission system and methods here are directed to wireless systems (see Imai et al. Fig. 22) and are thereby obvious to be implemented into any wireless system including a wireless microphone.

Regarding claim 14, depending on claim 13, its limitations are identical to the system (receiver) limitations of claim 10 and is therefore rejected under similar rationale. Note that the transmission system and methods here are directed to wireless systems (see Imai et al. Fig. 22) and thereby are obvious to be implemented into any wireless system including a wireless microphone.

Conclusion

5. The prior art made of record and not relied upon is considered pertinent to applicant's disclosure. Matsui et al. (US Patent 6,539,356), Serizawa (US Patent 6,101,464), Banba et al. (US 2004/0064310), Banba (US 2004/0008768), Banba et al. (JP 2002-330075).

Any inquiry concerning this communication or earlier communications from the examiner should be directed to FARZAD KAZEMINEZHAD whose telephone number is (571)270-5860. The examiner can normally be reached on M-F 8:30AM-5:00 PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Talivaldis I. Smits can be reached on (571)272-7628. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information

system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

/FK/

/Talivaldis Ivars Smits/ 7/6/2009
Primary Examiner, Art Unit 2626